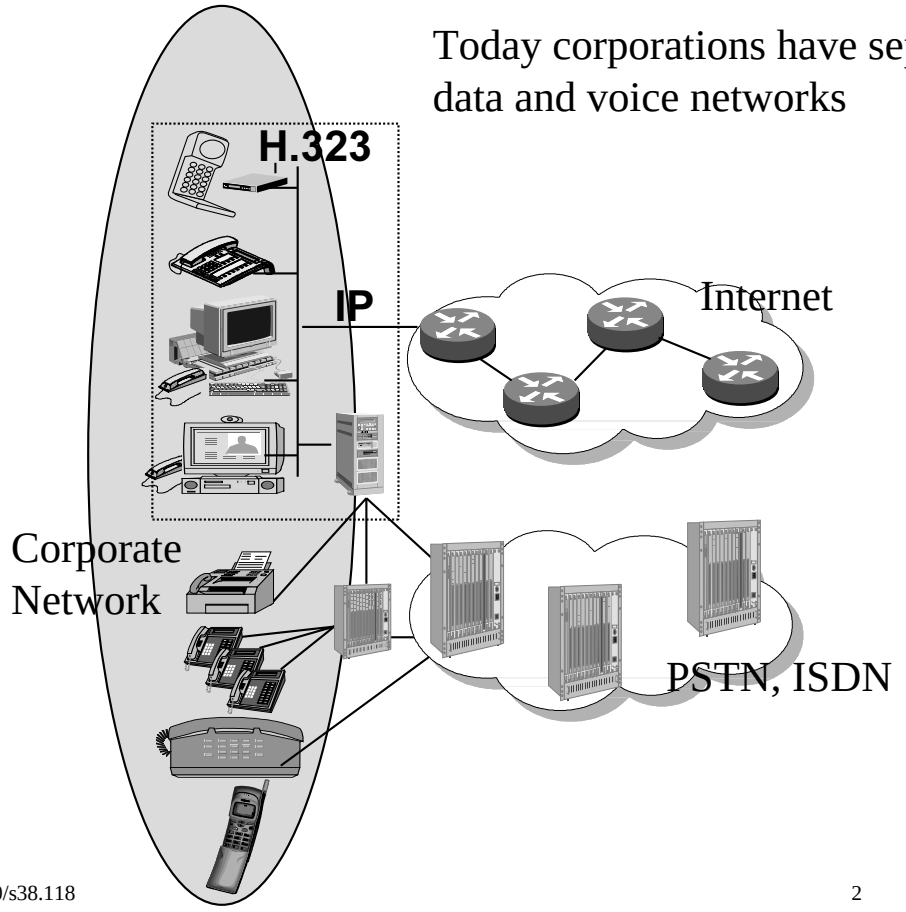
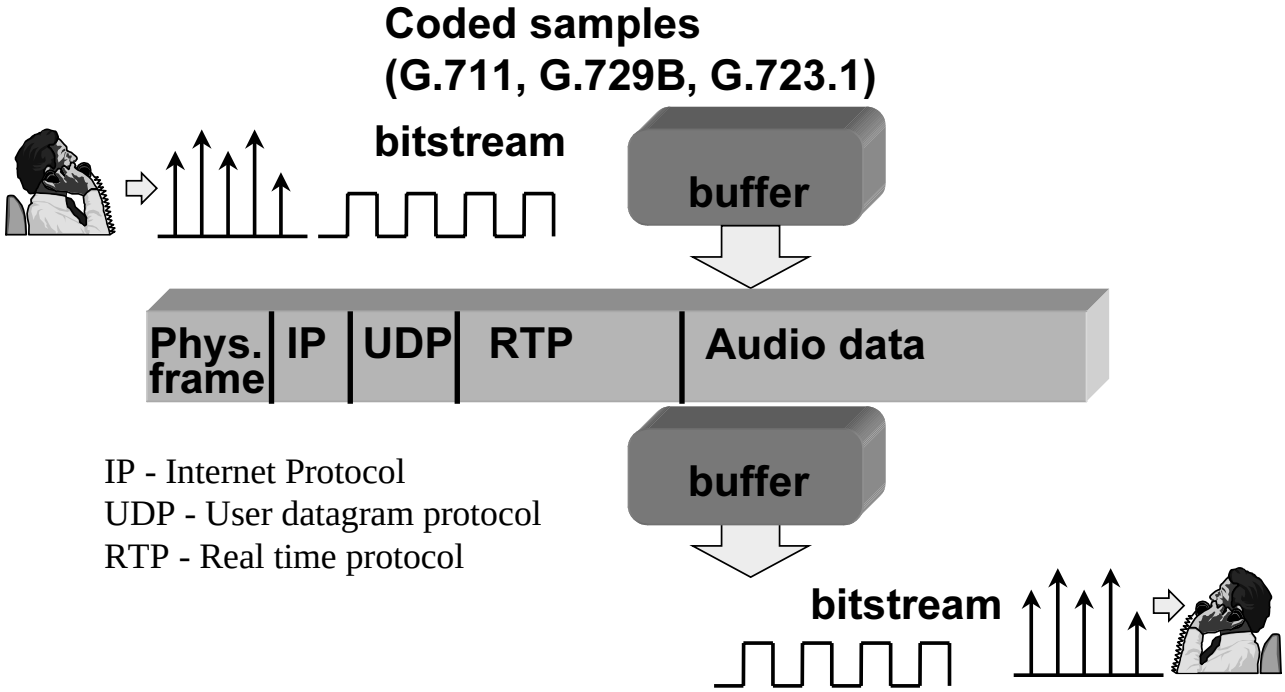
IP Telephony and Network Convergence

Raimo.Kantola@hut.fi

Today corporations have separate data and voice networks

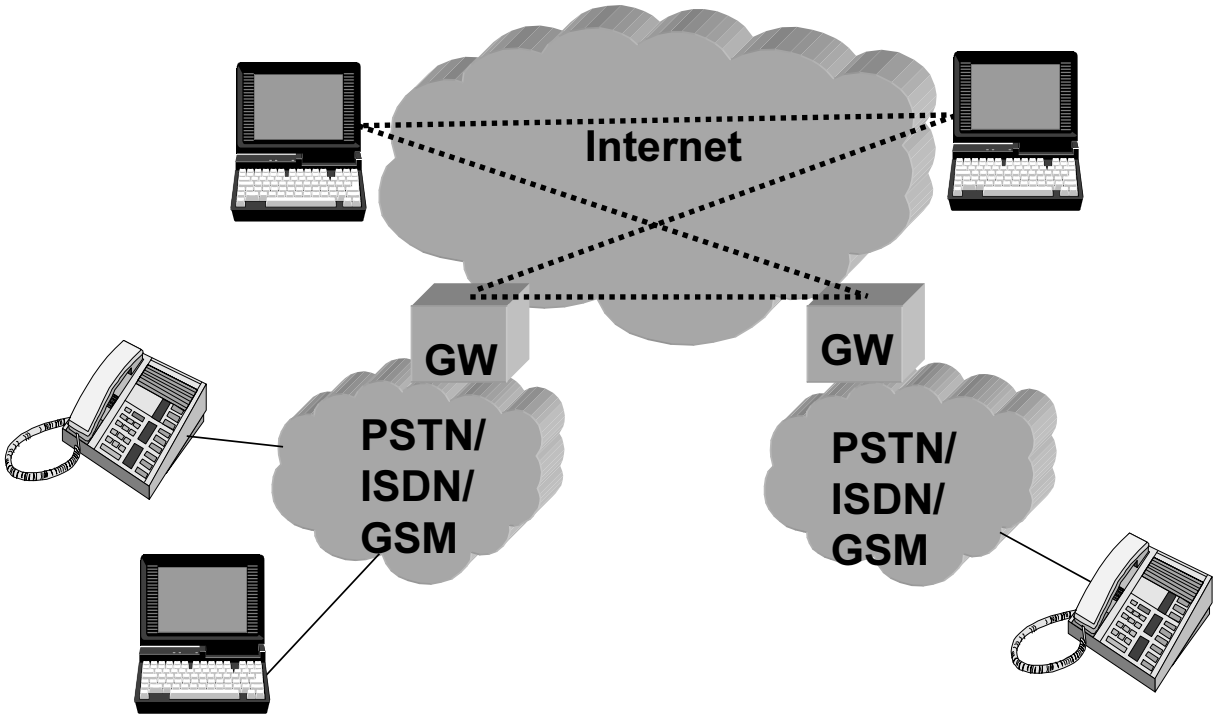


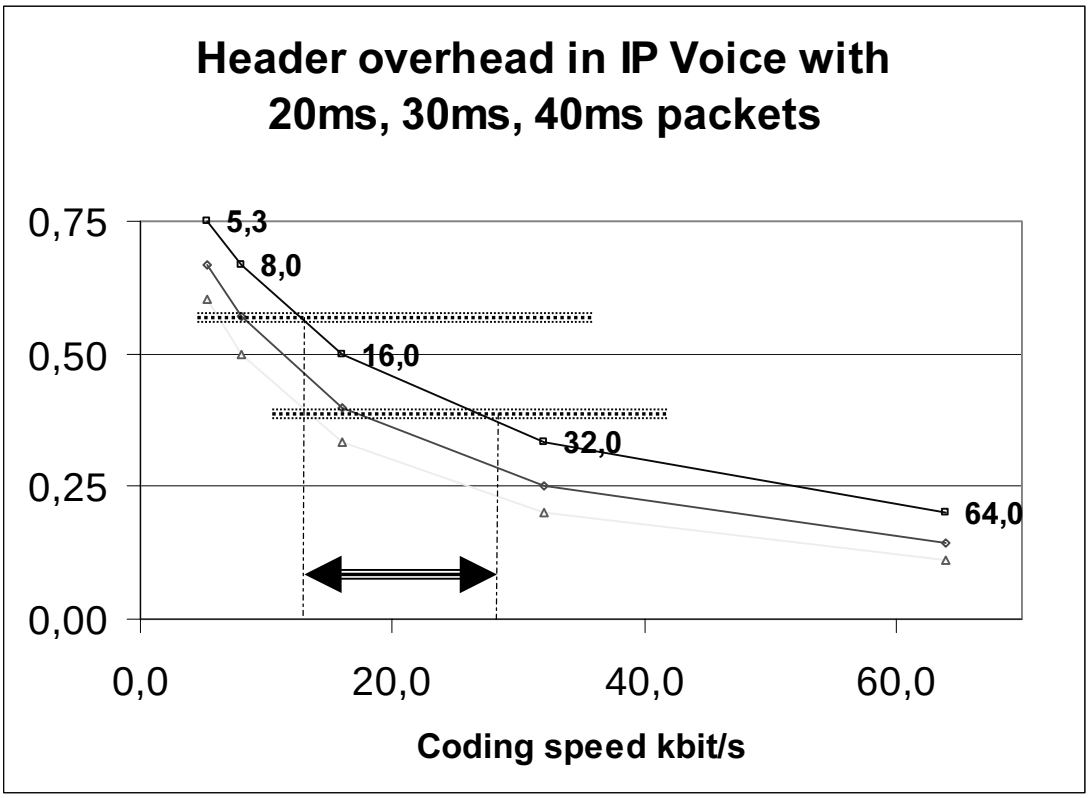
VoIP in action



IP - Internet Protocol
UDP - User datagram protocol
RTP - Real time protocol

Voice over IP





So, what about header overhead?

- It seems to make sense to choose voice coding speed around 15…30 kbit/s
- Bandwidth requirement can be reduced by header reduction in access and by silence detection in the backbone.
- IP still requires less than 64 kbit/s!

Why VoIP when ISDN/GSM works perfectly well?

Note: Voice still brings 90% of operator revenues!

- Maintaining two networks is expensive.
- Data traffic grows >30%/year, Voice ca. 5% and the volumes are approximately equal now. If this trend remains, in 2010 voice will be 10%, data will be 90%.
- Cost of transmission goes down very fast: xDSL, SDH, WDM - it is difficult to take full benefit of this trend using circuit switching: only one voice sample can be switched at a time: 8 bit sample vs e.g. 20 ms sample => 1 Gbit router is less expensive than 1 Gbit circuit switch.
- More processing can be pushed to terminals -> consumer market economics
- Convergence of Voice and Data can give service benefits.

Interoperability Issues

- Signaling and Call control
- Quality of Service

Phase 1 --->

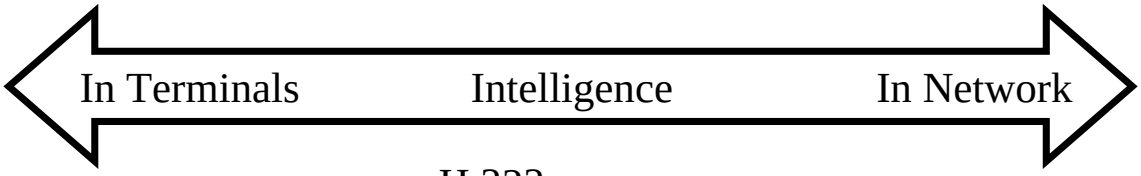
- Telephony Routing and addressing
  - Input Information gathering
  - Alternative routing over IP

Phase 2 -->

- Service Management in the hybrid network

Phase 3

Signalling alternatives



SIP
- ascii based
- devil in details
- also NNI coming
- Bakeoffs drive vendor interoperability

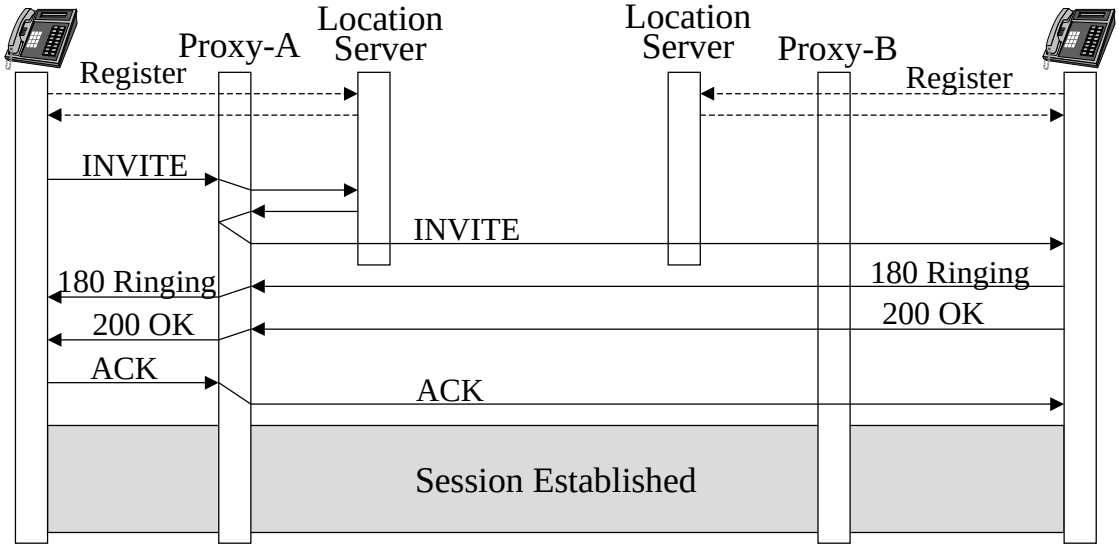
H.323
- Inherits ISDN
- complex
- still few services
- Vendor Interop has been announced

Megaco/H.248/MGCP
- newest
- seems to be quality spec.
- architecture holds promise
- Interoperability?

SIGTRAN(IETF) works on ISUP over SCTP over IP
- many (netheads) view this as an interim solution!

SIP - Session Initiation Protocol, H.323 - ITU-T specs for conferencing and voice over IP, Megaco - Media Gateway Control Protocol, ISUP - ISDN User Part, SCTP - Signalling Control Transport Protocol ("TCP optimised for signalling transport")

SIP call setup example



- Features: SIP is ASCII and similar to html/http
- "Location server" (includes DNS and) translates telephone numbers to IP address
- Proxy-B may need to be consulted. Instead of proxies, redirect servers may be used

How to provide SCN-like QoS over IP?

- Integrated Services (use RSVP to make reservations in routers for each call!) changes Routers into SCN-Exchange -like systems. Does not scale well.
- DiffServ or MPLS enchanced with QoS support
  - mark voice packets with higher than BE priority at ingress
  - priority queuing in transit nodes
  - How to prevent voice from blocking BE traffic?
  - How to do Service Management?
  - Voice packets have high overhead - how to minimize?
- Overprovisioning for voice

How is IP Telephony different from Circuit switched telephony?

IP Telephony

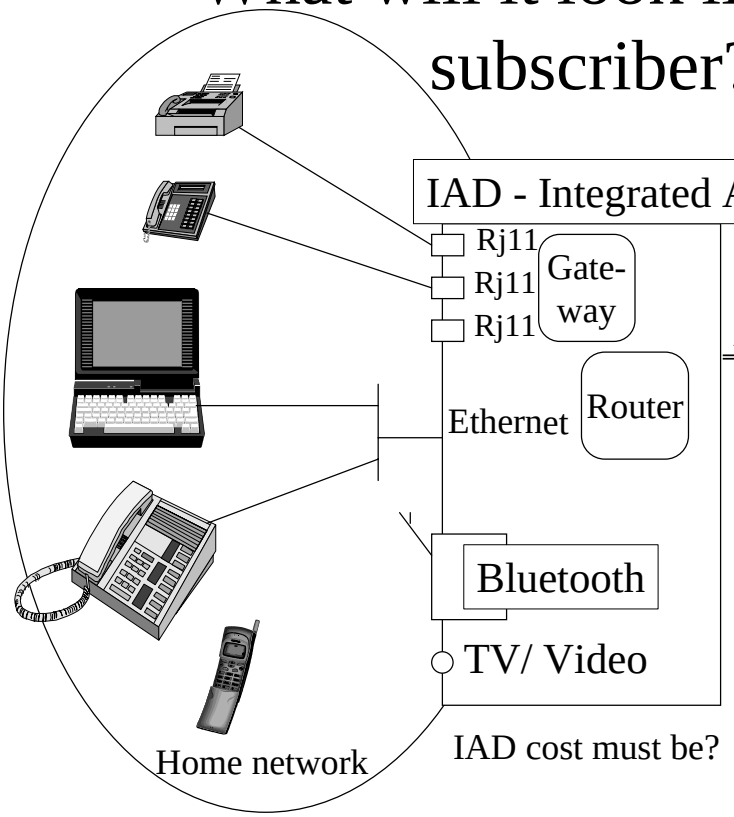
- Voice in 10…40 ms samples, Bits in a sample can be switched in parallel
- No single coding standard
- End-to-End delay is big challenge
- Terminals are intelligent - consumer market economics
- Call control is separate from voice path - first find out whether parties want and can talk, if yes, set-up the voice path

Circuit Telephony

- Voice sample = 8 bits
- A- and μ -law PCM voice standard
- Reference connection gives network design guidelines => end-to-end delay is under control
- Wireline telephones are dumb. Cellular phones are pretty smart
- Call control is tied to the voice path - IN is used to add service processing on the side.

Note: Using todays technology IP Telephony is not less expensive in replacement nor green field investments in Corporate networks!

What will it look like to the subscriber?



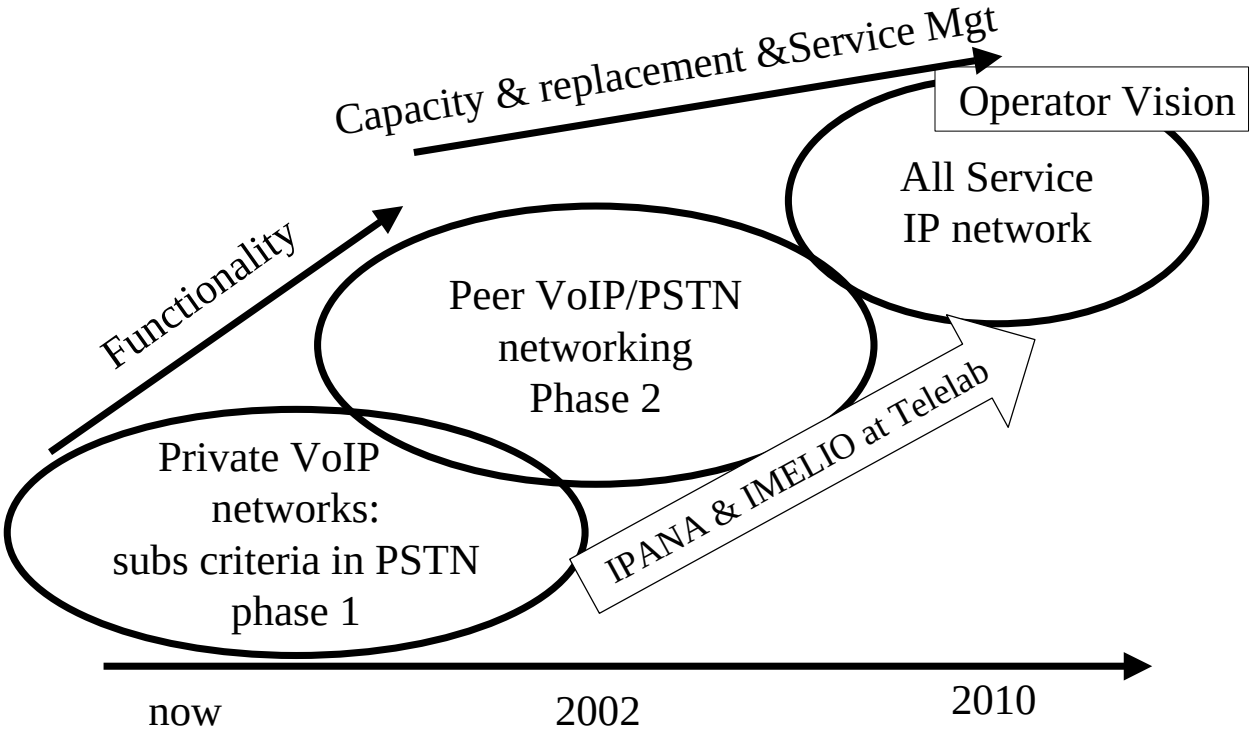
ADSL or VDSL

Or CATV

IAD will have voice gateway and Router functions

ADSL - Asymmetric Digital Subscriber Line
VDSL - Very high speed Digital Subscriber Line
CATV - Cable TV

Roadmap to the Future



Conclusions

- IP Telephony will become mainstream in the first 10 years of your professional lives
- Technology is not ready yet - more research and a lot of product development is needed
- ISDN/IN will not disappear overnight -> Interoperability of networks and services is key to convergence

What is convergence?

- Between data, voice and video services and networks
- Digital packet switching technology forms the basis for data, voice and video services
- All services have digital content
- Any network can be used to carry any service
- Any device (phone, PC, TV) can be used to access any service.